Amendments to the Claims:

This listing of claims will replace all prior versions and listing of claims in the application. In this response, claim 28 has been amended.

1. (Previously Presented) A method of reducing noise in an input signal, said input signal containing speech and noise related to each other by a signal to noise ratio, the method comprising the steps:

(1) detecting the presence and absence of speech;

(2) in the absence of speech, determining a noise magnitude spectral estimate $(|\hat{N}(f)|)$;

(3) in the presence of speech, comparing the magnitude spectrum of the input signal (|X(f)|) to the noise magnitude spectral estimate $(|\hat{N}(f)|)$;

(4) calculating an attenuation function (H(f)) from the magnitude spectrum of the input signal (|X(f)|) and the noise magnitude spectral estimate $(|\hat{N}(f)|)$, the attenuation function (H(f)) being dependent on the signal to noise ratio; and,

(5) modifying the input signal by the attenuation function (H(f)) to generate a noise reduced signal wherein there is no substantial modification to the input signal for very low and for very high signal to noise ratios.

2. (Previously Presented) A method as claimed in claim 1, further comprising the steps of:

(6) supplying the input signal to an amplification unit;

(7) providing the noise reduced signal to a compression circuit which generates a control signal for the amplification unit; and

(8) controlling the amplification unit with the control signal to modify the input signal to generate an output signal with compression and reduced noise.

3. (Previously Presented) A method as claimed in claim 2, wherein step (6) comprises subjecting the input signal to a main noise reduction algorithm to generate a

main noise reduced signal and providing the main noise reduced signal to the amplification unit.

4. (Previously Presented) A method as claimed in claim 3, wherein the main noise reduction algorithm comprises the method of claim 1.

5. (Previously Presented) A method as claimed in claim 3, wherein the main noise reduction algorithm is different from the method of claim 1.

6. (Previously Presented) A method as claimed in claim 25, wherein the attenuation function is calculated in accordance with the following equation:

$$H(f) = \left[\frac{|X(f)|^2 - \beta |\hat{N}(f)|^2}{|X(f)|^2} \right]^{\alpha}$$

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where $H(f)$ is the attenuation function, $|X(f)|$ is the magnitude spectrum of the input signal; $|\hat{N}(f)|$ is the noise magnitude spectral estimate, β is an oversubtraction factor and α is an attenuation rule, wherein α and β are selected to give a desired attenuation function.

7. (Original) A method as claimed in claim 6, wherein the oversubtraction factor β is varied as a function of the signal to noise ratio, with β being zero for high and low signal to noise ratios and with β being increased as the signal to noise ratio increases above zero to a maximum value at a predetermined signal to noise ratio and for higher signal to noise ratios β decreases to zero at a second predetermined signal to noise ratio greater than the first predetermined signal to noise ratio.

8. (Previously Presented) A method as claimed in claim 7, wherein the oversubtraction factor β is divided by a preemphasis function $P(f)$ to give a modified oversubtraction factor $\hat{\beta}(f)$, the preemphasis function being such as to reduce $\hat{\beta}(f)$ at high frequencies, and thereby reduce attenuation at high frequencies.

9. (Previously Presented) A method as claimed in claim 6, wherein the rate of change of the attenuation function (H(f)) is controlled to prevent abrupt and rapid changes in the attenuation function (H(f)).

10. (Previously Presented) A method as claimed in claim 6, wherein the attenuation function (H(f)) is calculated at successive time frames, and the attenuation function (H(f)) is calculated in accordance with the following equation:

$$G_n(f) = (1-\gamma)H(f) + \gamma G_{n-1}(f)$$

wherein $G_n(f)$ and $G_{n-1}(f)$ are the smoothed attenuation functions at the n'th and (n-1) 'th time frames, and γ is a forgetting factor.

11. (Original) A method as claimed in claim 10, wherein β is a function of perceptual distortion.

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12. (Previously Presented) A method as claimed in claim 1 which includes remotely turning noise suppression on and off.

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13. (Previously Presented) A method as claimed in claim 1 which includes automatically disabling noise reduction in the presence of very light noise or extremely adverse environments.

14. (Previously Presented) A method as claimed in claim 1 which includes detecting speech with a modified auto-correlation function.

15. (Original) A method as claimed in claim 14, wherein the auto-correlation function comprises:

(1) taking an input sample and separating it into short blocks and storing the blocks in correlation buffers;

(2) correlating the blocks with one another, to form partial correlations; and

(3) summing the partial correlations to obtain a final correlation.

16. (Original) A method a claimed in claim 15, wherein the method is carried out by digital signal processing and wherein the method includes using a Fast Fourier

Transform to generate the partial correlations and includes detection of voiced speech directly in the frequency domain.

17. (Previously Presented) A method as claimed in claim 1, wherein detecting the presence or absence of speech comprises:

(1) taking a block of the input signal and performing an auto-correlation on that block to form a correlated signal; and,

(2) checking the correlated signal for the presence of a periodic signal having a pitch corresponding to that for a desired audio signal.

18. (Original) A method as claimed in claim 17, wherein the auto-correlation is performed on a first block taken from the input signal, and a delayed block from the audio signal.

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19. (Original) A method as claimed in claim 18, wherein each block is subdivided into a plurality of shorter sections and the correlation comprises a correlation between pairs of the shorter sections to form partial correlations, and subsequently summing the partial correlations to obtain the correlated signal.

20. (Original) A method as claimed in claim 19, wherein an input signal is stored as a plurality of samples in a pair of correlation buffers, and the auto-correlation is performed on the signals in the buffers to determine the partial correlations, which partial correlations are summed and stored.

21. (Previously Presented) An apparatus, for reducing noise in an input signal, the apparatus including an input for receiving the input signal, the apparatus comprising:

(a) a compression circuit for receiving a compression control signal and generating an amplification control signal in response;

(b) an amplification unit for receiving the input signal and the amplification control signal and generating an output signal with compression and reduced noise; and,

(c) an auxiliary noise reduction unit connected to the input for generating an auxiliary noise reduced signal, the compression control signal being the auxiliary noise reduced signal.

22. (Previously Presented) An apparatus as claimed in claim 27, wherein the input signal contains speech and the main noise reduction unit comprises:

(1) a detector connected to said input and providing a detection signal indicative of the presence of speech;

(2) magnitude means for determining the magnitude spectrum of the input signal (|X(f)|), with both the detector and the magnitude means being connected to the input of the apparatus;

(3) spectral estimate means for generating a noise magnitude spectral estimate $\left(\left|\hat{N}(f)\right|\right)$ and being connected to the detector and to the input of the apparatus;

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(4) a noise filter calculation unit connected to the spectral estimate means and the magnitude means, for receiving the noise magnitude spectral estimate $\left(\left|\hat{N}(f)\right|\right)$ and magnitude spectrum of the input signal (|X(f)|) and calculating an attenuation function (H(f)); and,

(5) a multiplication unit coupled to the noise filter calculation unit and the input signal for producing the noise reduced signal.

23. (Previously Presented) An apparatus as claimed in claim 22, which includes a frequency transform means connected between said input and both of the magnitude means and the spectral estimate means for transforming the signal into the frequency domain to provide a transformed signal (X(f)) wherein the magnitude means determines the magnitude spectrum (|X(f)|) from the transformed signal (X(f)), and wherein the spectral estimate means determines the noise spectral estimate $\left(\left|\hat{N}(f)\right|\right)$ from the transformed signal (X(f)) in the absence of speech, the apparatus further including inverse frequency transform means for receiving a transformed noise reduced signal from the multiplication unit, the inverse frequency transform means providing the noise reduced signal.

24. (Previously Presented) An apparatus as claimed in claim 23, wherein the noise filter calculation unit determines the square of the speech magnitude spectral estimate by subtracting the square of the noise magnitude spectral estimate from the square of the magnitude spectrum of the input signal and wherein the noise filter calculation unit calculates the attenuation function (H(f)), as a function of frequency, in accordance with the following equation:

$$H(f) = \left[\frac{|X(f)|^2 - \beta |\hat{N}(f)|^2}{|X(f)|^2} \right]^{\alpha}$$

where f denotes frequency, $H(f)$ is the attenuation function, $|X(f)|$ is the magnitude spectrum of the input audio signal; $|\hat{N}(f)|$ is the noise magnitude spectral estimate, β is an oversubtraction factor and α is an attenuation rule, wherein α and β are selected to give a desired attenuation function.

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25. (Previously Presented) A method as claimed in claim 1, wherein the square of the speech magnitude spectral estimate $(|\hat{S}(f)|)$ is determined by subtracting the square of the noise magnitude spectral estimate $(|\hat{N}(f)|)$ from the square of the magnitude spectrum of the input signal (|X(f)|).

26. (Previously Presented) A method as claimed in claim 2, wherein step (6) comprises applying steps (1) to (5) to the input signal prior to supplying the input signal to the amplification unit.

27. (Previously Presented) An apparatus as claimed in claim 21, wherein the apparatus further comprises a main noise reduction unit connected to the input for generating a noise reduced signal and supplying the noise reduced signal to the amplification unit in place of the input signal.

28. (Currently Amended) An apparatus as claimed in claim 27, wherein the main noise reduction unit and the auxiliary noise reduction unit ~~comprise a single~~ employ the same noise reduction algorithm~~unit~~.

29. (Previously Presented) An apparatus as claimed in claim 27, wherein the auxiliary noise reduction unit is different from the main noise reduction unit.

30. (Previously Presented) An apparatus as claimed in claim 22, wherein the input signal contains speech and noise related to each other by a signal to noise ratio and the noise filter calculation unit produces the noise reduced signal in dependence upon the signal to noise ratio, wherein there is no substantial modification to the input signal for very low and for very high signal to noise ratios.

31. (Previously Presented) A method of reducing noise in an input signal, said input signal containing speech and noise related to each other by a signal to noise ratio, the method comprising the steps:

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(1) detecting the presence and absence of speech;

(2) in the absence of speech, determining a noise magnitude spectral estimate $\left(\left|\hat{N}(f)\right|\right)$;

(3) in the presence of speech, comparing the magnitude spectrum of the input signal (|X(f)|) to the noise magnitude spectral estimate $\left(\left|\hat{N}(f)\right|\right)$;

(4) calculating an attenuation function (H(f)) from the magnitude spectrum of the input signal (|X(f)|) and the noise magnitude spectral estimate $\left(\left|\hat{N}(f)\right|\right)$, the attenuation function (H(f)) being dependent on the signal to noise ratio; and,

(5) modifying the input signal by the attenuation function (H(f)) to generate a noise reduced signal wherein there is no substantial modification to the input signal for very low and for very high signal to noise ratios and wherein the amount of attenuation provided by the attenuation function is increased as the signal to noise ratio increases above zero to a maximum value at a predetermined signal to noise ratio and for higher signal to noise ratios the amount of attenuation provided by the attenuation function decreases to zero at a second predetermined signal to noise ratio greater than the first predetermined signal to noise ratio.

32. (Previously Presented) An apparatus, for reducing noise in an input signal containing speech and noise related to each other by a signal to noise ratio, the apparatus including an input for receiving the input signal, the apparatus comprising:

(a) a compression circuit for receiving a compression control signal and generating an amplification control signal in response;

(b) an amplification unit for receiving the input signal and the amplification control signal and generating an output signal with compression and reduced noise; and,

(c) an auxiliary noise reduction unit connected to the input for generating an auxiliary noise reduced signal, the compression control signal being the auxiliary noise reduced signal,

wherein the auxiliary noise reduction unit generates the auxiliary noise reduced signal in dependence upon the signal to noise ratio, wherein there is no substantial modification to the input signal for very low and for very high signal to noise ratios.

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33. (Previously Presented) An apparatus, for reducing noise in an input signal containing speech and noise related to each other by a signal to noise ratio, the apparatus including an input for receiving the input signal, the apparatus comprising:

(a) a compression circuit for receiving a compression control signal and generating an amplification control signal in response;

(b) an amplification unit for receiving the input signal and the amplification control signal and generating an output signal with compression and reduced noise; and,

(c) an auxiliary noise reduction unit connected to the input for generating an auxiliary noise reduced signal, the compression control signal being the auxiliary noise reduced signal,

wherein the auxiliary noise reduction unit generates the auxiliary noise reduced signal according to an attenuation function in dependence upon the signal to noise ratio, wherein there is no substantial modification to the input signal for very low and for very high signal to noise ratios and wherein the amount of attenuation provided by the

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attenuation function is increased as the signal to noise ratio increases above zero to a maximum value at a predetermined signal to noise ratio and for higher signal to noise ratios the amount of attenuation provided by the attenuation function decreases to zero at a second predetermined signal to noise ratio greater than the first predetermined signal to noise ratio.